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AUDIO NON-LINEARITY: an initial appraisal of a double comb-filter method of measurement

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AUDIO NON-LINEARITY: AN INITIAL APPRAISAL OF A DOUBLE COMB-FILTER METHOD OF MEASUREMENT

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Summary

A new objective method of measuring non-linear distortion of sound signals is described. It uses a test signal comprised of two pseudo-random noise signals whose spectrum occupies most of the audio band. Distortion is measured by using two comb-filters to separate the test-noise signal from distortion products.

The method gives better correlation with subjective test results than the conventional total-harmonic distortion method and also reduces the inter-channel cross-talk that can arise when testing transmission circuits. It is likely that the cost of equipment for the new distortion-measuring technique could be less than that of total-harmonic distortion equipment.

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AUDIO NON-LINEARITY: AN INITIAL APPRAISAL OF A DOUBLE COMB-FILTER METHOD OF MEASUREMENT

AUDIO NON-LINEARITY: AN INITIAL APPRAISAL OF A DOUBLE COMB-FILTER METHOD OF MEASUREMENT

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1. Introduction

This Report describes recent experimental studies of a new technique for measuring sound-signal non-linearity. This technique is a variant of the single comb-filter method described in Reference 1 and it will be referred to as the double comb-filter method. It is instrumentally less complex than the single comb-filter method as it obviates the need for a frequency shifter. As most of the instrumentation comprises digital shift-registers, the design can take advantage of future cost reductions in these devices produced by large scale integration technology, and may eventually be cheaper to produce than total harmonic distortion equipment.

In an earlier Report[1] it was established that the single comb-filter method gave significantly better correlation between measurement and subjective assessment than was provided by a routine measurement of total-harmonic distortion. It should be emphasised, however, that the comparison was not a general one, in that different correlation results would be obtained if total-harmonic distortion measurements were made at a higher or lower test signal level than that used for routine test purposes within the BBC. Nevertheless, the results of other researchers[2,3] who have studied the total-harmonic distortion method using test-signal levels significantly below test circuit overload, support the conclusion that a total-harmonic distortion test is not the best means of estimating subjective impairment.

Another disadvantage of the total-harmonic distortion test is that the single high-level tone test signal can cause inter-channel cross-talk, particularly on sound-signal circuits derived by FDM techniques on a carrier system. Such tests usually have to be restricted to infrequent short bursts to minimise the cross-talk problem and further constraints, which may be imposed in due course by the UK Post Office, will increase the operational difficulties incurred. With the proposed new form of test signal however, the inter-channel cross-talk is likely to be much less serious because of the dispersed-energy nature of the signal. This advantage has been confirmed by measurements made by the UK Post Office using the signal employed in the earlier single comb-filter technique and it would be expected to apply for the variant on this method described in this Report.

Earlier subjective investigations to assist the appraisal of new non-linear distortion-measuring techniques[1,4] were restricted to programme impairment produced by a gain non-linearity which was independent of signal frequency. This Report describes how the effects of frequency-dependent non-linearity were subsequently investigated in an initial subjective study of somewhat limited scope. Using these new subjective data, a fuller comparison is drawn between the subjective/objective correlation obtained with the double comb-filter method and that obtained by a routine total harmonic distortion test.

2. Description of double comb-filter method

The test signal used with the double comb-filter method has a spectrum made up of the addition of two comb structures of different frequency. As each comb defines a harmonically related spectrum, those non-linearity products which arise by intermodulation between one comb spectrum and the other will fall in the gaps of the test-signal spectrum. Two comb-filters, aligned with the test signal spectrum, will therefore remove the test-signal components and enable the remaining distortion products to be measured.

The test signal can conveniently be generated by using

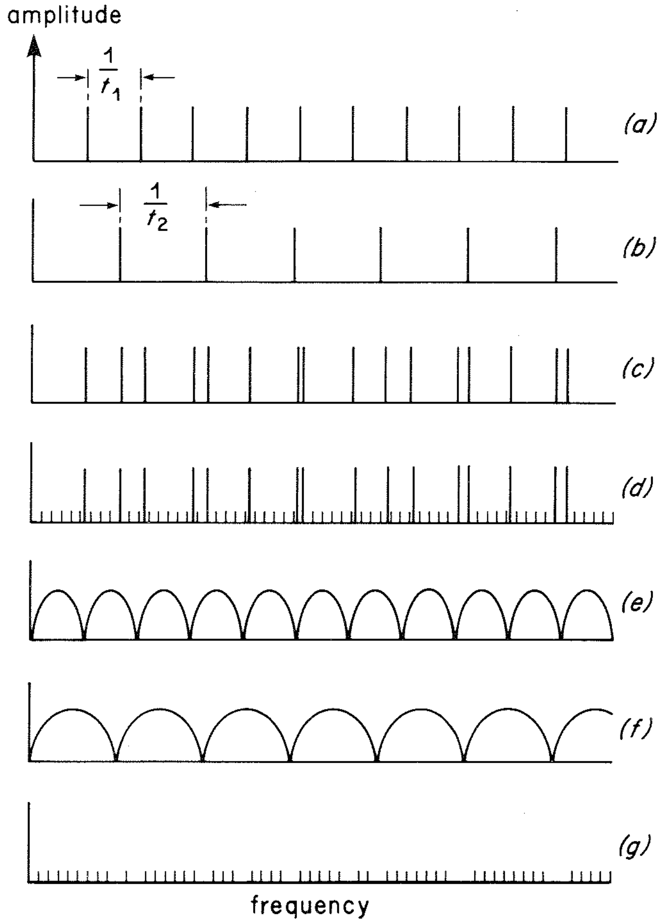

Fig. 1 - Typical spectra

t_1 = repetition period of m-sequence G_1

t_2 = repetition period of m-sequence G_2

(*a*) m-sequence G_1 (*b*) m-sequence G_2 (*c*) Test signal

(*d*) Signal from system under test (*e*) First comb response

(*f*) Second comb response (*g*) Output from comb-filters

two maximal-length pseudo-random binary sequence generators (m-sequences[5]). The spectrum of a binary signal produced by an m-sequence generator consists of harmonics of the sequence repetition frequency which are all of substantially equal amplitude, as illustrated in Fig. 1(*a*) and (*b*), provided the required test-signal bandwidth is less than one tenth of the sequence clock-frequency. When two such signals are added, the test-signal spectrum illustrated in Fig. 1(*c*) can be obtained. When non-linearity products are generated by this signal in a system or circuit under test, the output spectrum will be as illustrated in Fig. 1(*d*). A simple comb response of the form illustrated in 1(*e*) may be used to remove one of the m-sequence signals, and a second filter (see Fig. 1(*f*)) in tandem to remove the other m-sequence, leaving non-linearity products as illustrated in Fig. 1(*g*)

Not all non-linearity products can be recovered; those occupying the same frequency slots as the test-signal are lost, but it is expected from past experience[1] that measurement of the available distortion products will be sufficient to give good correlation with the associated subjective impairment.

3. Experimental details

3.1. Apparatus

Fig. 2 is a simplified block diagram of the arrangement used for generating the test signal. G_1 and G_2 are m-sequence generators whose cycle times are controlled by two crystal oscillators. Circuit details of G_1 and G_2 are given in the Appendix, Fig. 1A. A scaling amplifier adds

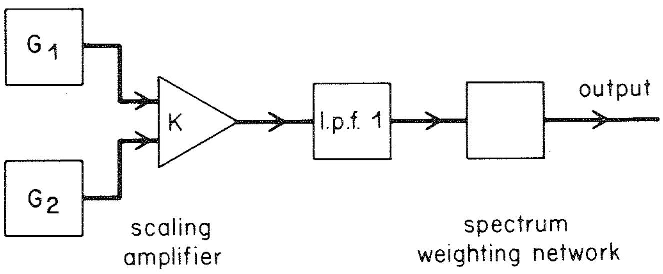

Fig. 2 - The test-signal generator

the two output signals in the ratio K to 1, where K is optimised by experiment. The low-pass filter l.p.f. 1 restricts the bandwidth of the combined signal to 15 kHz, and the resulting waveform is noise-like. In order to facilitate a study of the importance of the shape of the test-signal spectrum, the output from the low-pass filter may be processed by a spectrum-weighting network which can introduce an adjustable amount of frequency-dependent loss and dispersion.

Fig. 3 shows a simplified block diagram of the test-signal analyser. Digital processing is used because of its stability and relative cheapness in providing the required performance. Circuit details of the delta-modulation coder (analogue-to-digital converter), delta-modulation subtractors,* and delta-modulation decoder (digital-to-analogue converter) are given in the Appendix, Fig. A2. The output signal from the system under test is applied to the delta-modulation coder. The delta modulator is an analogue-to-digital converter which provides a 1-bit output signal at a rate of approximately 4 Mbit/s. This form of digital signal was used as it is less expensive to produce than a conventional high accuracy (12-bit) pulse-code modulation (p.c.m.) signal. However, recent work on delta-modulation to p.c.m. conversion of high-quality sound signals[6] indicates that delta-modulation combined with p.c.m. comb filters may be a more attractive arrangement in the future. Each delay is produced by a digital shift-register and is part of a comb-filter which operates by subtracting from an input signal a delayed version of itself. In Fig. 3, shift-register delay t_1 is equal to the cycle time of the m-sequence produced by G_1, and similarly for t_2 and G_2. The delta-modulation decoder converts the digital signal into analogue form by integration. The attenuation peaks of the comb-filter responses formed by the shift-register arrangements in Fig. 3 are maintained in accurate alignment with the test-signal 'teeth' by using a crystal-controlled clock-pulse generator; for clarity, Fig. 3 does not show clock-signal paths.

The gain of the comb-filtered path is set so that, at a signal frequency at which minimum loss is produced by the comb-filters, the output level from the 'measure' path is equal to that from the 'set 0 dB' path.

* Acknowledgement is made to M. Weston, BBC Research Department, for the subtractor design.

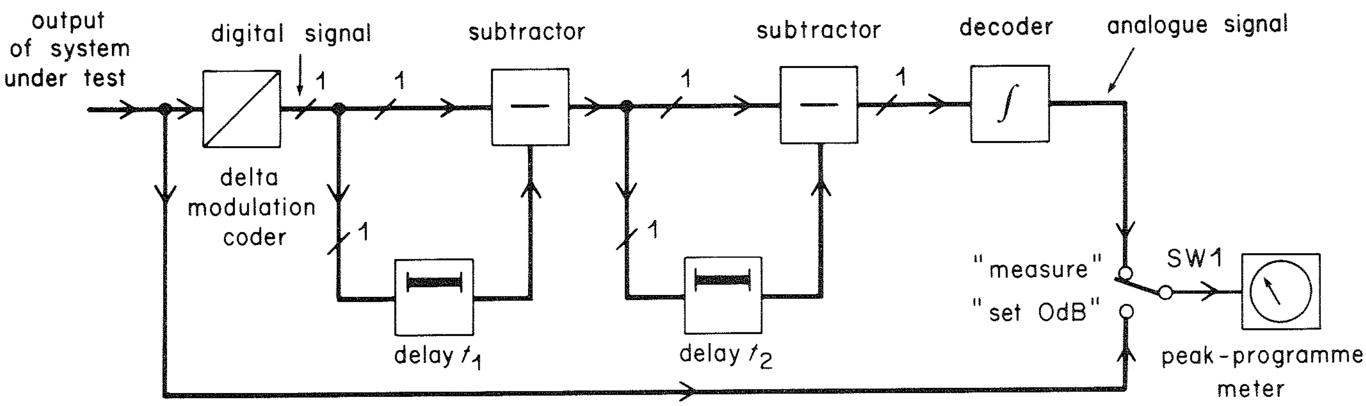

Fig. 3 - The test-signal analyser

A maximum separation figure of 54 dB was measured when the test-signal generator output was connected directly to the input of the measurement system depicted by Fig. 3. This separation is later shown to be at least 17 dB greater than that for which distortion would be audible. The separation was limited by quantisation noise produced by the delta-modulation codec: an equivalent separation would probably be obtained if a 12-bit p.c.m. codec operating at a 32 kHz sampling rate were used instead.

The clock frequency chosen for the delta-modulation was a compromise between keeping the quantisation noise of the delta modulation codec at a low level (a higher clock frequency would have produced lower quantisation noise) and keeping to a minimum the required number of shift-register elements (the required shift-register length is proportional to clock frequency).

In the experimental equipment, MOS shift-register devices were used and these had a specified maximum data rate of 10 Mbit/s. Emitter-coupled logic circuits were used in the delta-modulation coder and decoder in order to produce pulse waveforms with well-matched rise and fall times. This matching is essential in order to maintain low harmonic distortion within the measuring equipment, since the coder and decoder operate by analogue integration of pulse waveforms. An alternative solution would require the pulse shape in the decoder to be adjusted for minimum signal distortion at the output of the decoder, but this solution was not used as it would require more complex circuitry.

3.2. Method of measuring noise-separation

Using the arrangement shown in Fig. 3 the switch SW1 is set to the 'set 0 dB' position and the output test-signal level from the system under test is adjusted until a reading of 0 dB is indicated by the peak-programme meter (PPM).[8] Noise-separation in dB is then indicated by the PPM when SW1 is set to 'measure'. It should be noted that this reading is a measure of (signal + distortion)/(distortion), and although not a true separation figure, its use has proved to be satisfactory in past experiments[1] with subjective data obtained for circuits whose non-linearity was not a function of signal frequency (within the audio-frequency range).

In order to increase the scope of the earlier subjective-objective correlation studies it was decided to conduct first a limited series of subjective tests, as described in Section 4, to obtain data relating the quality impairment of programme to the applied signal level for two circuits which exhibited a frequency-dependent non-linearity.

4. An assessment of programme quality impairment with non-linearity more pronounced at high signal frequencies

4.1. Introduction

The purpose of the tests was to obtain subjective estimates of programme quality impairment caused by two sound programme circuits for a selection of applied signal

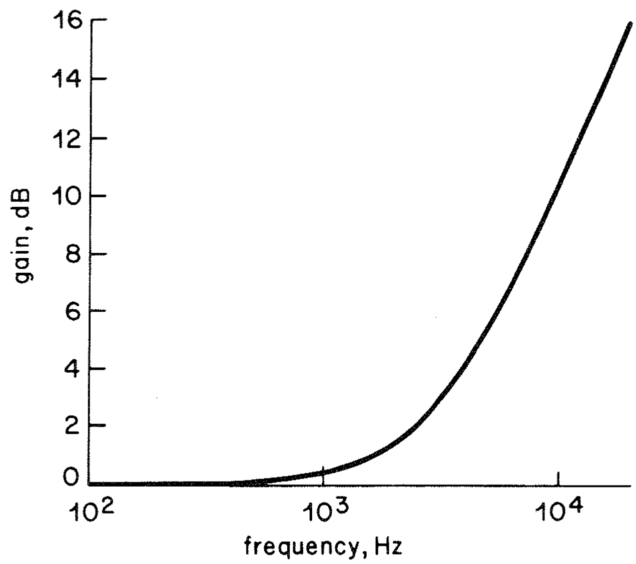

Fig. 4 - 50 μs pre-emphasis characteristic

levels. A 50 μs pre-emphasis network having the amplitude-frequency characteristic shown in Fig. 4 was included at the input of each circuit under test. This network caused high-frequency signals to be more susceptible than low-frequency signals to non-linear distortion caused by over-driving the circuit under test, thus giving a condition representative of a transmission system using pre-emphasis. A network with the complimentary characteristic was included at its output to provide an overall flat amplitude-frequency characteristic.

No amplitude compression was applied to the signal and the gain of the programme source was set to ensure that programme peaks registered no more than scale mark '6' on a peak-programme meter (at least one peak indicated exactly '6') at a point in the circuit where the line-up tone indicated scale mark '4' ('6' corresponds to a signal level 8 dB higher than indicated by '4').

The tests explored impairment levels in the range indicated by the six-point scale shown in Table 1. This scale was used to enable accurate comparisons to be made with subjective data obtained in the earlier investigation.[4] In CCIR Report 623 (1974), a five-point scale is given for assessing impairment of sound, and this scale will be adopted by the BBC in new studies. As a first approximation, the linear relationship $A_5 = 5{\cdot}8 - 0{\cdot}8A_6$ can be used to transform a six-point grade A_6 as used in this Report into a CCIR five-point grade A_5.

TABLE 1

The Six-Point Subjective Impairment Scale

Grade	Impairment
1	Imperceptible
2	Just perceptible
3	Definitely perceptible but not disturbing
4	Somewhat objectionable
5	Definitely objectionable
6	Unusable

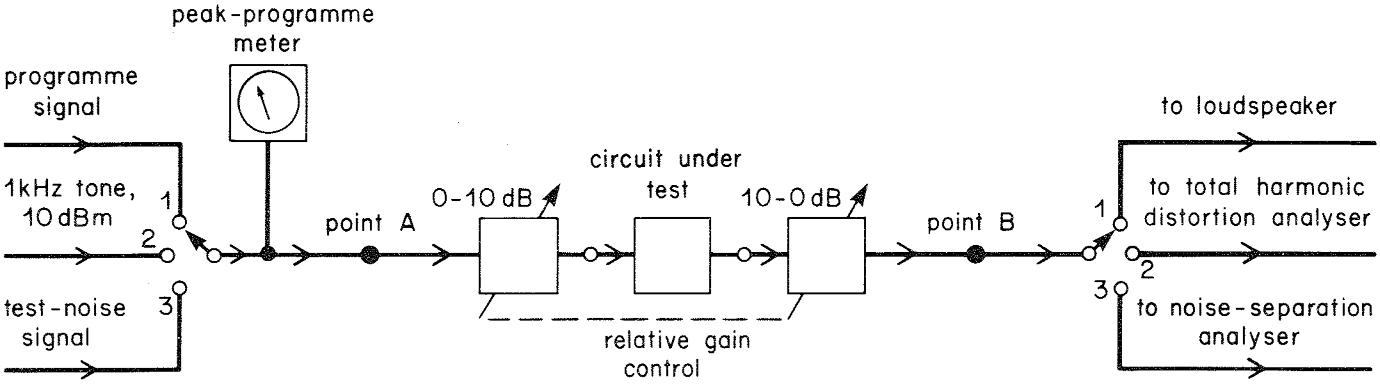

Fig. 5 - Test arrangement

4.2. Choice of test-programme excerpts and test-circuits

A solo piano item was selected from a library of recorded test-excerpts as it was found to be subjectively more sensitive to distortion in the test-circuits than the other excerpts.

A companding system was used in the tape recording and reproduction of the test-programme so that recording noise was less likely to mask the distortion produced by the test-circuits. In addition, the programme signals went through the record-replay process only once, i.e. a 'master' recording was used for the tests.

The two test-circuits used were a transistorised line-receiving amplifier, BBC type AM7/4, and an integrated-circuit operational amplifier type 741. These were two of the four circuits used in the earlier study[4] and their use here allowed subjective data obtained in the earlier study to be combined with results from the present one.

4.3. Test procedure

Since, in contrast with the earlier tests,[4] only master recordings were used, it was possible for the listeners to grade each item as it was presented, without the need for each condition to be preceded by a nominally unimpaired item. Twenty-four conditions were presented during the test and in these only 11 different degrees of impairment were presented; these were repeated in a random fashion so that the consistency of each assessment could be estimated. One of the 11 was produced by by-passing the non-linear circuit, to provide a reference condition.

The tests were carried out in a listening room having a mean mid-band reverberation time of approximately 0·3 seconds and a volume of 85 cu. metres using a high-quality monitoring loudspeaker BBC type number LS5/5. Each test condition was judged by six listeners experienced in assessing sound quality. The listeners sat at a mean distance of 2·5 metres from the loudspeaker; the reproduced sound level measured on axis at this distance was approximately 85 dBA, obtained with a sound-level meter to IEC Publication Number 123, set to the 'slow' time-constant. The background acoustic noise level was 25 dBA.

Fig. 5 shows the test arrangement used. The 'relative gain' ganged attenuators enabled the operating level of the circuit under test to be varied about a reference condition defined as 0 dB. This reference was arranged by adjusting the gain of the circuit under test so that with a +10 dBm 1 kHz tone applied to point A in Fig. 5 the total harmonic distortion measured at point B was —37 dB. This figure of —37 dB is somewhat arbitrary, but was chosen as it has in the past been taken as a tolerance limit for sound-programme circuits.

The +10 dBm tone corresponds to the level used in the BBC for routine total harmonic distortion checks on sound-programme circuits and is 2 dB greater than normal programme peaks as indicated on a PPM.

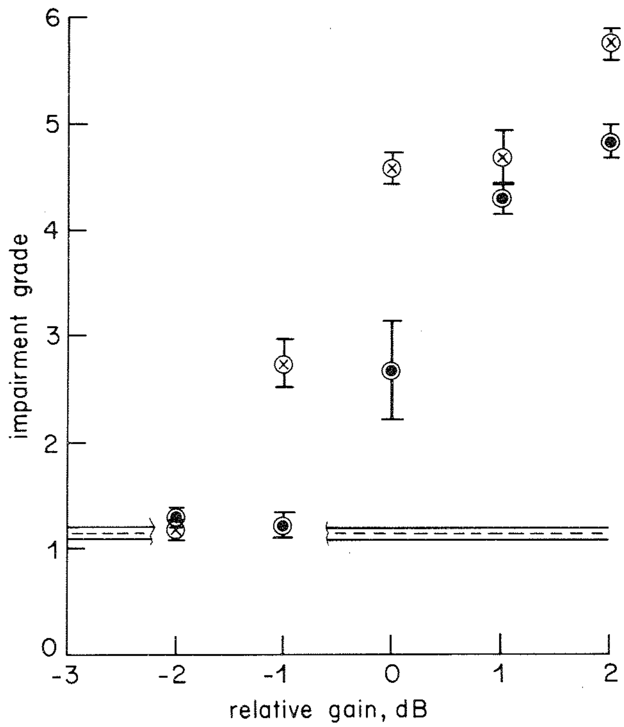

Fig. 6 - Average impairment grade versus relative gain for 'Solo Piano'

⊗ 741 ◉ AM7/4 I twice standard error with pre- and de-emphasis

reference condition including twice standard error

The test results obtained related subjective estimates of programme-quality impairment to settings of the relative gain control and these results are presented in Section 4.4.

4.4. Subjective data obtained

Mean, and standard error of the mean were calculated for the results of each test condition. Fig. 6 shows a plot of the results. It should be stated that the standard error of each mean grade given in Fig. 6 was arrived at in the following way. Each listener was asked to grade each degree of impairment twice on average during the test session as a check on the repeatability of his assessment. The average of his grading for each degree of impairment was then taken. The standard error of these six mean gradings was then calculated, on the basis of six listeners, in the usual fashion. It is of interest (see Fig. 6) that the mean grading for the reference condition was 1·15 with a standard error of 0·05. This shows that the background noise and distortion of the sound reproduction system did not seriously influence the test results. Listeners' ability to repeat assessments was in general good, though sometimes the two assessments by a given listener of one degree of impairment were different by one grade. This is reflected in an increased standard error, and may be an indication of the difficulty in judging a particular test condition. In order that the standard errors may be compared, the results of the earlier subjective tests[4] in which the same two circuits were used, but without pre- and de-emphasis networks, are presented again in Fig. 7. The test programme used for the results in Fig. 7 was the one found to be most critical in those tests and was an excerpt of male speech in which

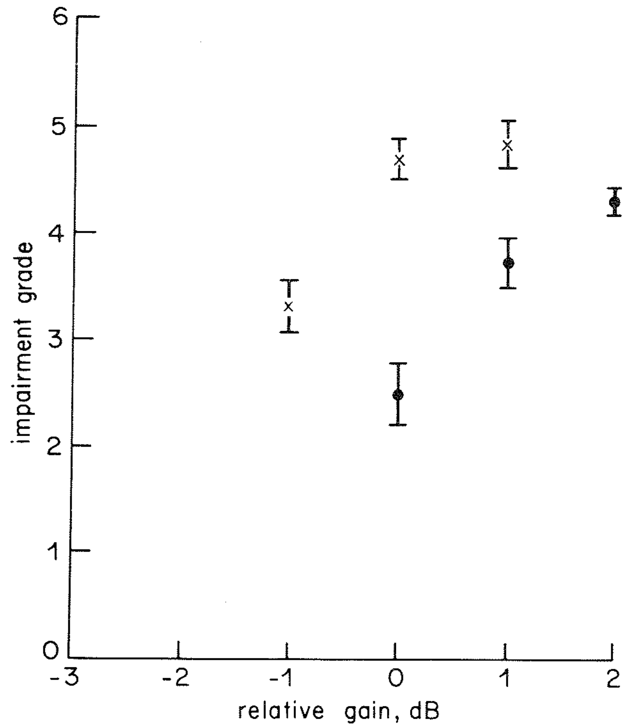

Fig. 7 - Average impairment grade versus relative gain for 'Male Speech'

× 741 ● AM7/4 I twice standard error
without pre- and de-emphasis

Harvard sentences were spoken: these sentences had been designed to be phonetically balanced.

(When programme items are chosen for subjective tests, one usually begins by examining a set which is representative of typical broadcast material. But it is usually found that one of the programme items is more susceptible to a particular type of distortion, e.g. a given amount of amplifier overload will produce the greatest quality impairment with one particular programme item. It is therefore necessary to ensure that an objective test for non-linearity should provide results which give good correlation with those obtained with a critical test programme item, even though the most critical programme may be different for different types of non-linearity.)

The data given in Figs. 6 and 7, therefore, enable the results obtained using the double comb-filter method of measurement to be related to the subjective appraisal of the circuits. The results of this study are given in Section 5.

5. Subjective-objective correlation with the double comb-filter method

5.1. Introduction

Measurements of noise-separation by the double comb-filter method, and also measurements of total harmonic distortion were made with the two test-circuits with and without the 50 μs pre- and de-emphasis circuits. The results of these measurements were plotted against relative gain.

The plots of objective non-linearity measurement against relative gain were then matched with plots of subjective impairment against relative gain and a graph of subjective impairment against objective distortion was drawn. Ideally, this graph would be a straight line for each non-linear circuit, and a perfect result would be obtained if one line were sufficient to describe all non-linear circuits. In practice this does not happen, but fairly straight lines can usually be obtained by a variety of distortion-measurement methods, for a certain number of test circuits. The best correlation is judged to be obtained by a test method which gives the closest grouping of plots along a line of subjective impairment against objective distortion.

Certain parameters in the double comb-filter method were known to affect the resulting subjective-objective correlation and experiments were therefore conducted to find the optimum parameter values. These were determined by plotting graphs of the type mentioned in the previous paragraph for various parameter values. The results of this optimisation procedure are presented in the next Section.

5.2. Optimum parameter values for the double comb-filter method

The parameters under study were: test-signal power, scaling amplifier gain K (see Fig. 2), test-signal base-

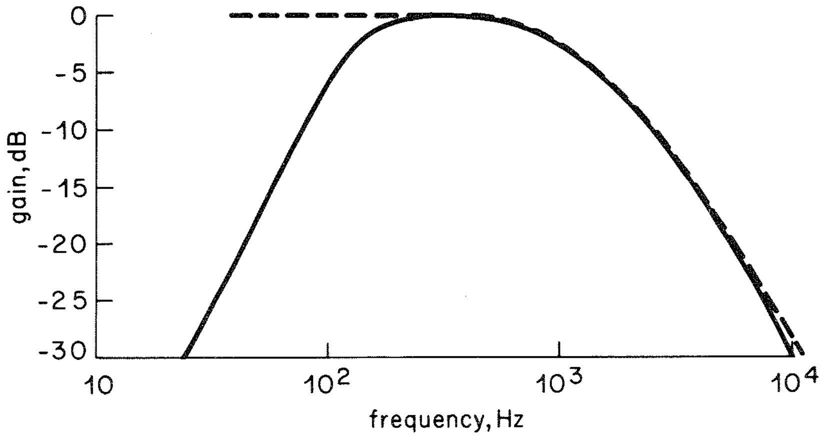

Fig. 8 - Test signal frequency-shaping characteristic

frequencies, overall spectral weighting of the test-signal, and spectral weighting of the distortion signal obtained from the comb-filters.

The best value of test-signal power was found to be 3 dBm, measured at point A in Fig. 5. This signal power gives a PPM reading of '6' which is also the nominal peak-programme-signal reading.

When amplifier gain K was optimum the spectrum of the test signal consisted of spectral lines which were all of equal power.

This form of test signal gave the maximum yield of distortion power, as would be expected from theory (it was not certain before the experiments whether improved subjective correlation might be obtained by a different peak-to-mean ratio of the test-signal from that given by a value of K which produced equal spectral-line powers).

Test-signal base-frequencies of 152·6 Hz and 109·8 Hz were found to give best subjective agreement.* These were produced by m-sequence lengths of 2047 and 4095 respectively; further details for these sequence generators are given in the Appendix Fig. A1.

Frequency-dependent weighting of the test-signal was found to be important, as it helped to reduce the spread in impairment attributable to a noise separation figure, taking into account both flat and frequency-dependent non-linear effects. The experimentally derived frequency-shaping characteristic is shown by the pecked line in Fig. 8. However, the characteristic shown by the full line in Fig. 8 can be obtained by connecting a CCIR average programme-weighting network[9] in series with a 50 μs de-emphasis network. This combination of 'standard' networks was used in the final version of the non-linearity test equipment as it gave equally good subjective agreement. It is interesting to note that this characteristic (full line in Fig. 8) agrees closely with plots of average programme spectra produced by the British Post Office in recent measurements[10] of the power loading produced by broadcast sound signals.

* The ratio of these two frequencies is the important parameter. The precise and rather odd frequency values result from the digital shift register increments used in the comb-filters. The use of other increments would have been instrumentally more difficult.

It was found to be advantageous to modify the phase-frequency response of the test-signal weighting network by including four single-section all-pass networks in tandem each having break frequencies of 700 Hz. This processing increased the PPM peak test-signal indication by approximately 1 dB for a given test-noise power.

No advantage was found in shaping the spectrum of the separated distortion signal output from the comb-filters.

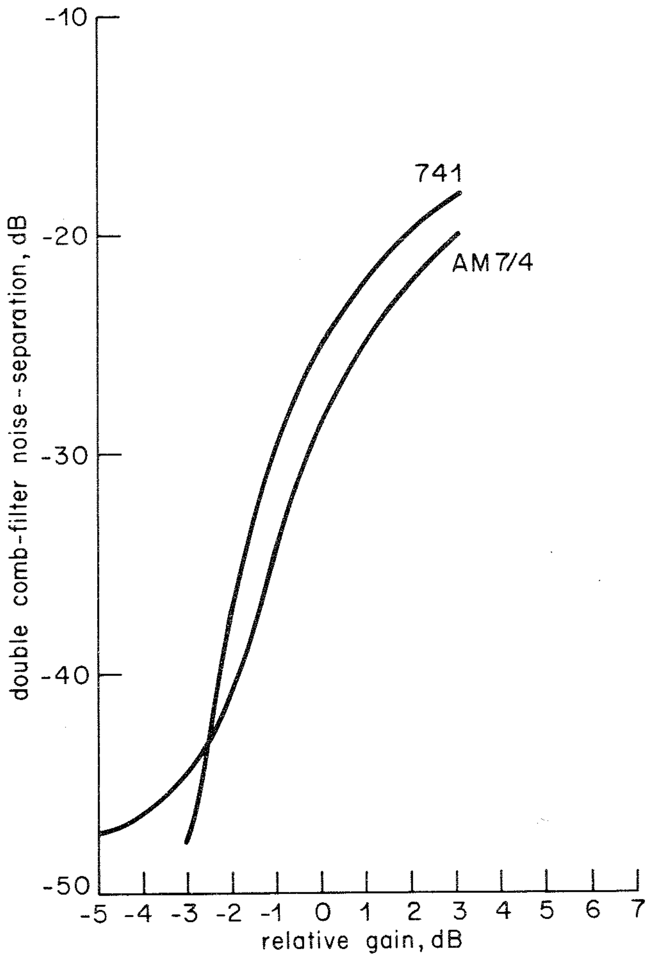

Fig. 9 - Double comb-filter noise-separation versus relative gain for test circuits without pre- and de-emphasis

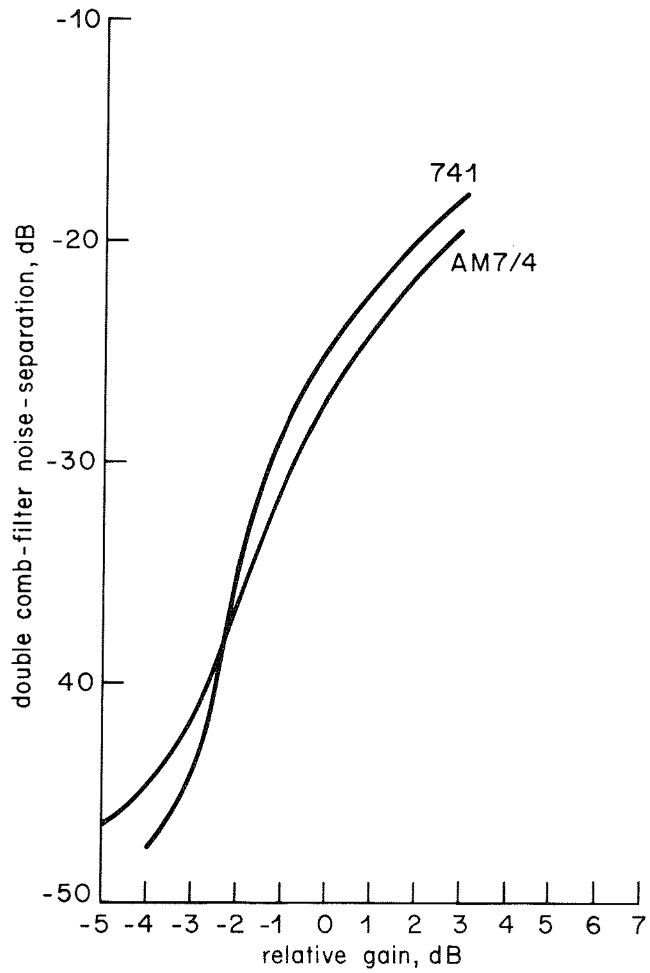

Fig. 10 - Double comb-filter noise-separation versus relative gain for test circuits with pre- and de-emphasis

Using the optimum arrangement described above, measurements of noise-separation against relative gain were made and the results are shown in Figs. 9 and 10. Measurements were also made of the total harmonic distortion using a test-frequency of 1 kHz (and test power of 10 dBm at point A in Fig. 5) and these results are plotted in Figs. 11 and 12.

These four Figures were then used in conjunction with Figs. 6 and 7 in deciding which of these two non-linearity test methods gave results which provided best subjective agreement.

5.3. A comparison of subjective-objective agreement obtained with total harmonic distortion measurements and with double comb-filter noise-separation measurements

The subjective measurements of distortion shown in Figs. 6 and 7 were combined (see Section 5.1) with the objective ones shown in Figs. 9 and 10, and 11 and 12, to give Figs. 13 and 14.

Fig. 13 shows the measured spread in noise-separation against subjective impairment for two circuits, with and also without 50 μs pre- and de-emphasis. Similarly, Fig. 14 shows the corresponding measured spread for total harmonic distortion.

Taking grade 2·5 as a reference point for the comparison, Fig. 14 shows that the spread in total harmonic distortion is at least 23 dB; it is apparent from the data shown that the routine total harmonic distortion method gives a very poor degree of correlation for different types of non-linear circuit.

Fig. 13 however, shows by the close grouping of plots (spread approximately 6 dB at grade 2·5) that the noise-separation test gives much better correlation between subjective and objective measurement for different types of non-linear circuits than does the total harmonic distortion method.

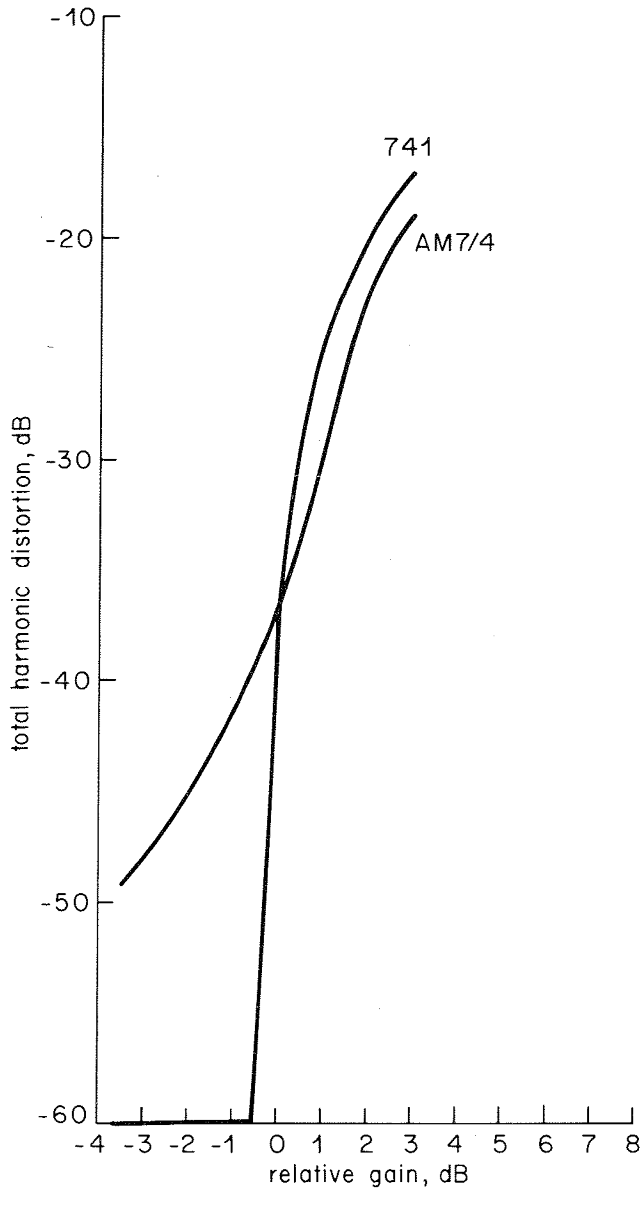

Fig. 11 - Total harmonic distortion versus relative gain for test circuits without pre- and de-emphasis

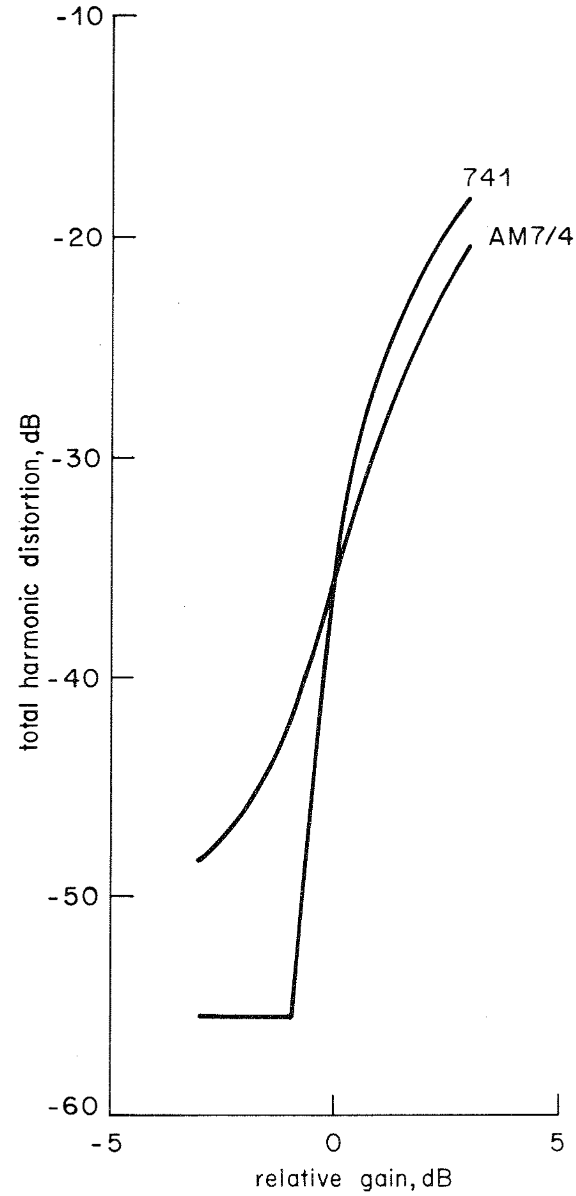

Fig. 12 - Total harmonic distortion versus relative gain for test circuits with pre- and de-emphasis

5.4. A limited comparison of subjective-objective agreement obtained with noise separation measured by single and double comb-filter methods

This Section describes a limited comparison between the single comb-filter method[1] and the double comb-filter method. It is limited in the sense that it applies only to two circuits, an operational amplifier type 741, and a line receiving amplifier type AM7/4, and with one critical test programme, a male speech excerpt. Fig. 15 shows the results of this comparison; the two non-linear circuits used were chosen as they gave results with a spread which was fairly representative of the spread obtained with the four circuits used initially.[1]

It can be seen that both noise-separation methods give plots with a gradient of approximately 3 dB per grade, and with spreads of 5 dB at grade 2·5. The double comb-filter method is, however, a more sensitive objective method,

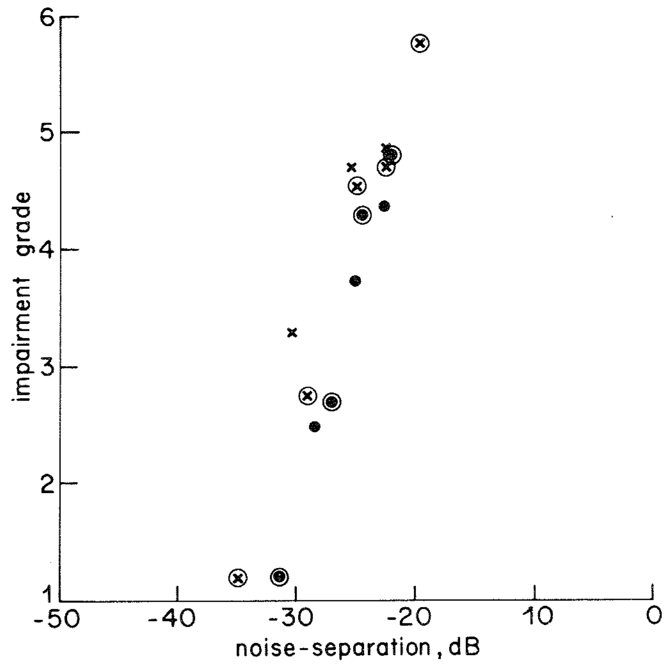

Fig. 13 - Average impairment grade versus double comb-filter noise-separation

× 741 ● AM7/4 ⊗ ◉ with pre- and de-emphasis

since for a given subjective impairment grade, the noise-separation is approximately 7 dB less than with the single comb-filter method. It may therefore be possible to either (a) use lower performance comb-filters with the double comb-filter method than with the single comb-filter

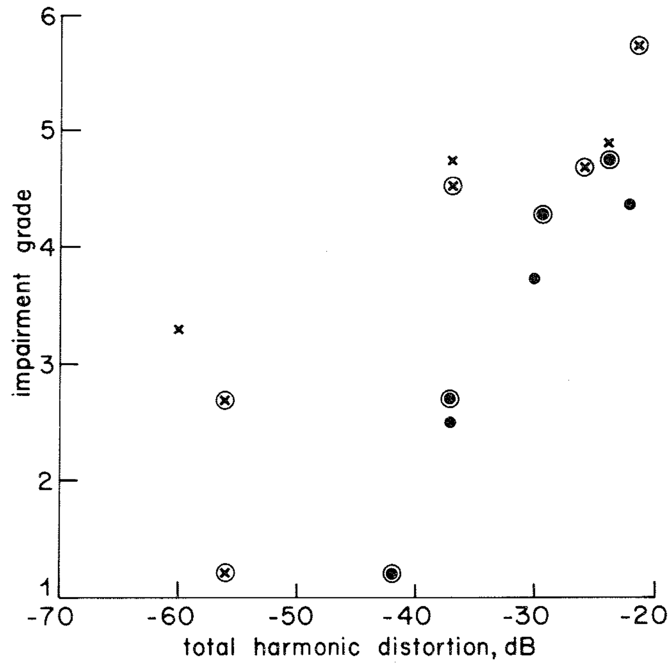

Fig. 14 - Average impairment grade versus total harmonic distortion

× 741 ● AM7/4 ⊗ ◉ with pre- and de-emphasis

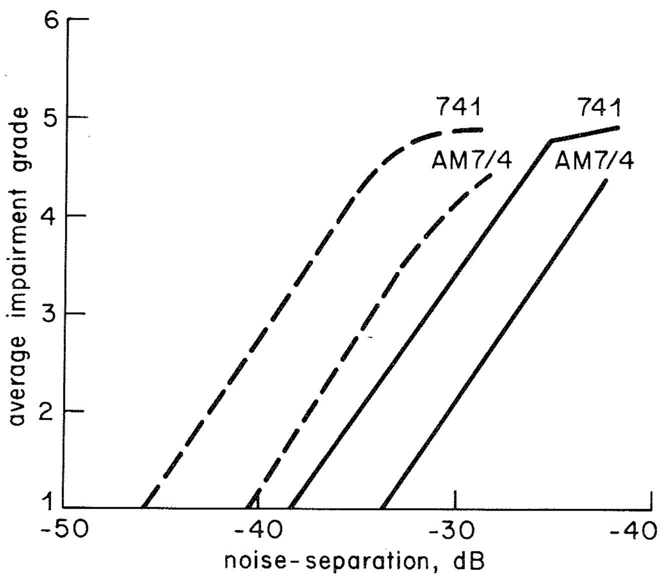

Fig. 15 - A comparison of double and single comb-filter methods

— — — single comb-filter method
——— double comb-filter method

method, or (b) maintain comb-filter performance and make use of the lower threshold of objective distortion measurement. Option (b) may be of interest if the noise-separation method is used for detecting variations in very low level 'background' distortion produced by equipment, as for example in type-approval testing. The maximum noise-separation obtained using delta-modulation comb-filters in the double comb-filter method was 54 dB, i.e. at least 17 dB greater than that for which distortion would be audible.

6. Conclusions

The double comb-filter method of non-linearity distortion measurement provides a degree of correlation between subjective and objective assessments of programme impairment which is much better than that provided by the routine total harmonic distortion method. Tests applied to circuits whose non-linearity was independent of signal frequency indicate that this correlation is at least as good as was earlier obtained with the single comb-filter method. In addition, the double comb-filter method has been studied using circuits with frequency dependent non-linearities and an equally good degree of correlation has been obtained.

The double comb-filter method is instrumentally less complex than the single comb-filter method and, as its instrumentation consists mainly of digital shift-register devices, it is to be expected that the future cost of instrumenting the method will fall. This should occur when suitable large-scale integrated circuits become available; the present trend is one of producing devices with smaller cost per shift-register element. With the advent of such devices it is possible that the cost of instrumenting the double comb-filter method may be less than that of instrumenting the total harmonic distortion method.

7. Recommendations

It is recommended that operational field trials be conducted to assess the usefulness of the double comb-filter method with a wider range of test-circuits than has so far been investigated. Its application to both routine testing and fault location should be studied.

It is also recommended that in the design of future equipment for the double comb-filter method, consideration should be given to employing a delta-modulation-to-p.c.m. converter,[6,7] as p.c.m. comb-filtering would permit a significant reduction in the required number of shift-register elements. Moreover, a delta-modulation-to-p.c.m. converter may be cheaper to produce than an equivalent high-accuracy conventional analogue-to-p.c.m. encoder.

8. References

1. BELCHER, R.A. 1976. Audio non-linearity: a comb-filter method for measuring distortion. BBC Research Department Report No. 1976/12.

2. SHORTER, D.E.L. 1950. The influence of high-order products on non-linear distortion. *Electronic Engineering*, 1950, **22**, 4, pp. 152 – 3.

3. WIGAN, E.R. 1961. New distortion criterion. *Electronic Technology*, 1961, **38**, 4 and 5, pp. 128 – 137, 163 – 174.

4. BELCHER, R.A. 1974. An experimental investigation of test-noise signals for the measurement of non-linear distortion of sound signals. BBC Research Department Report No. 1974/2.

5. GOLOMB, S.W. 1967. Shift register sequences. Holden Day, San Francisco, 1967.

6. BBC UK Patent Application No. 36211/76.

7. BBC UK Patent Application No. 47945/76.

8. British Standard Specification 4297:1968.

9. CCIR Geneva, 1974, V, XII, Report 497-1.

10. CCIR Geneva, 1974, V, XII, Report 491-1. Further information to be proposed at final meeting of study period 1974 – 1978.

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Appendix

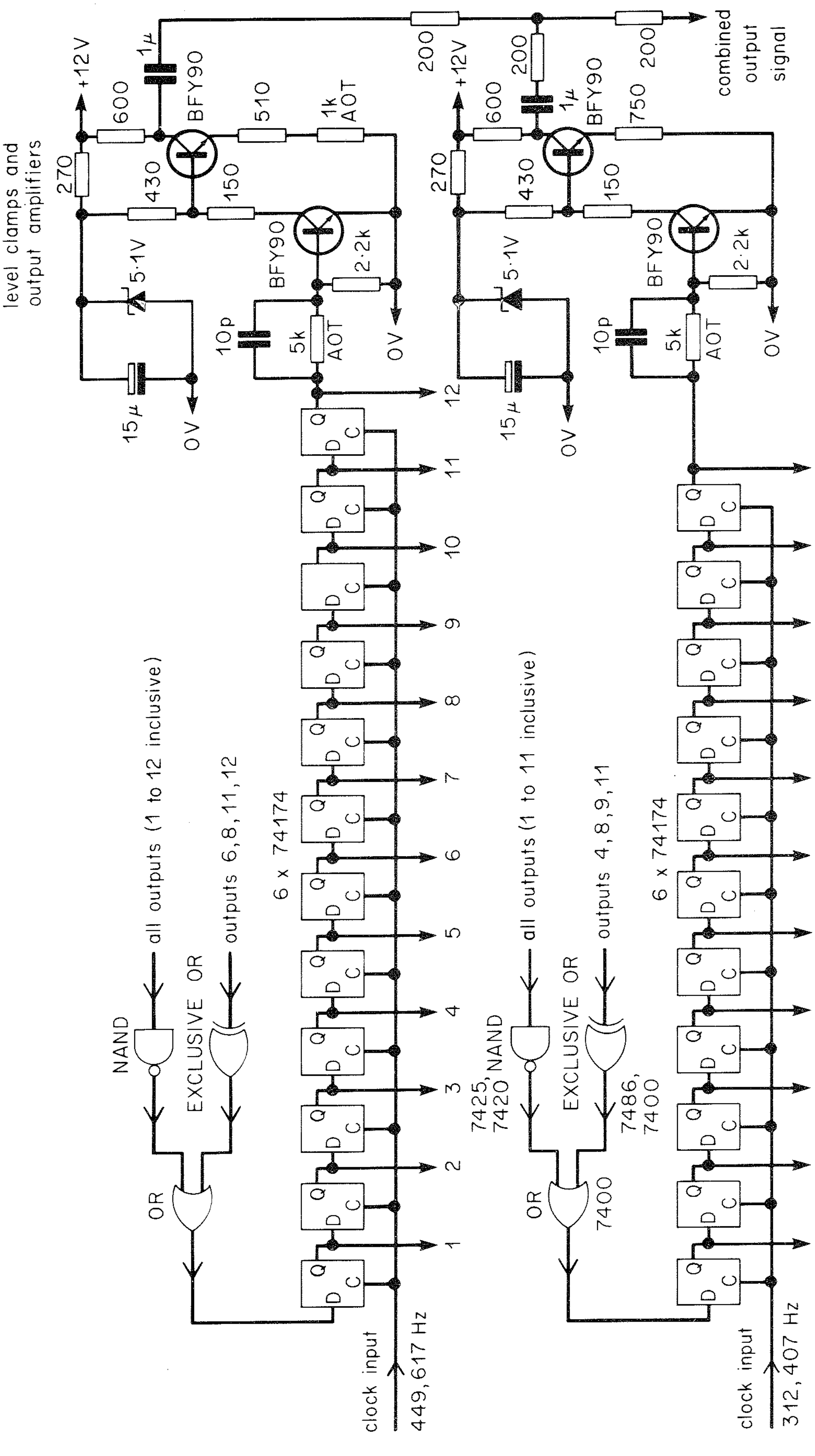
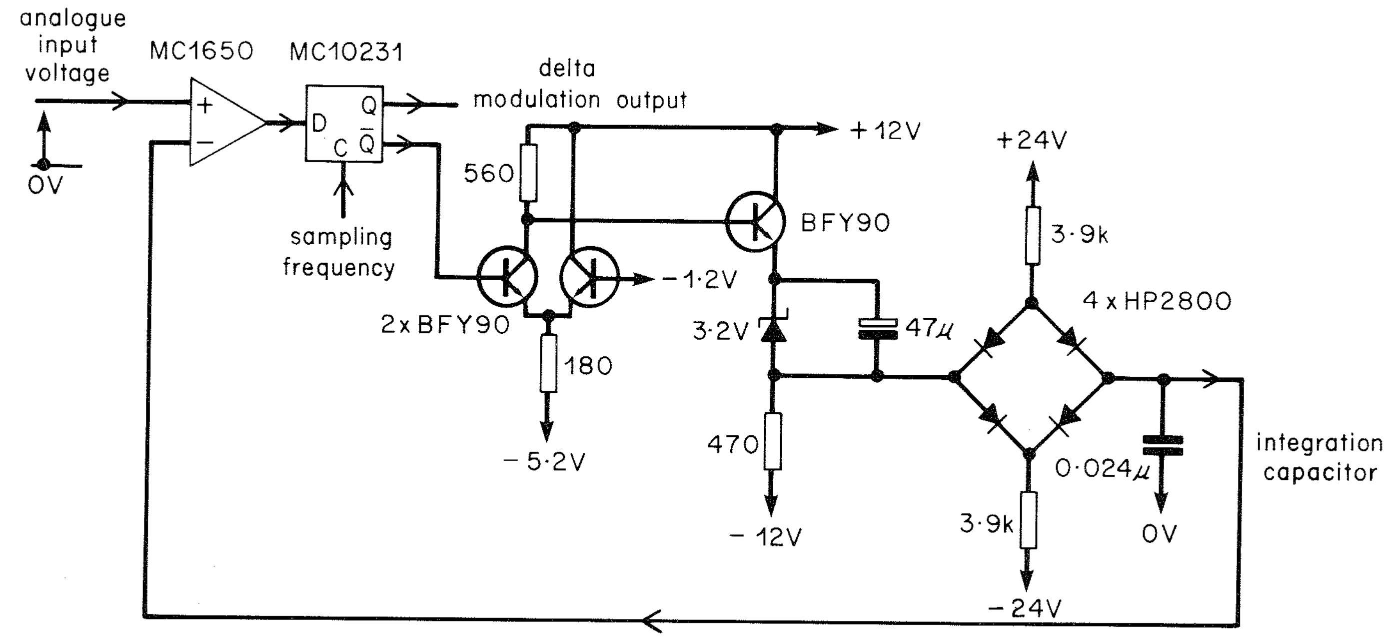
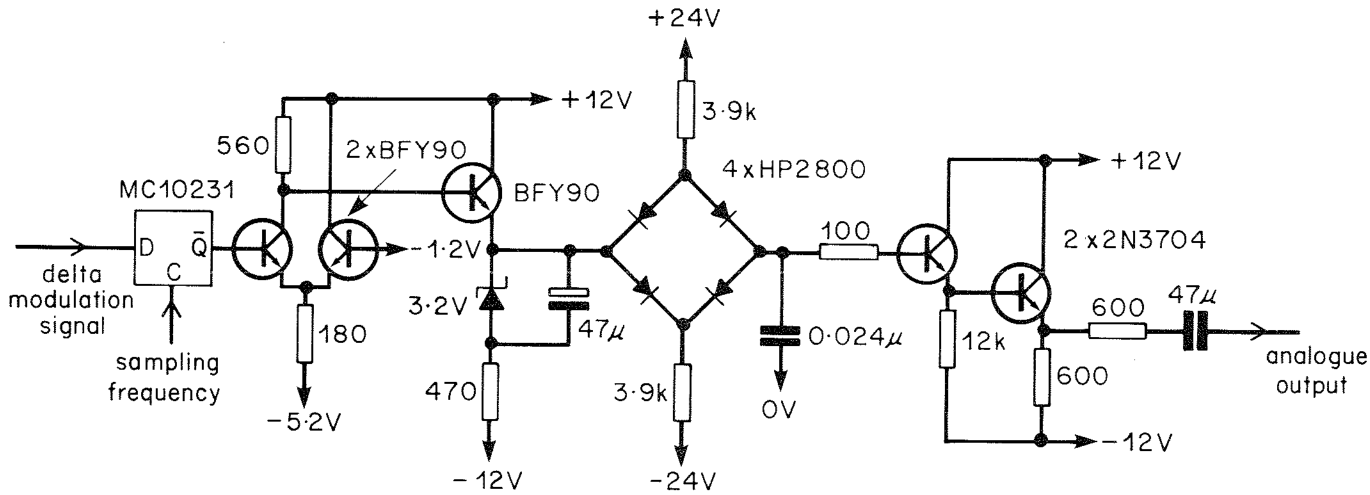


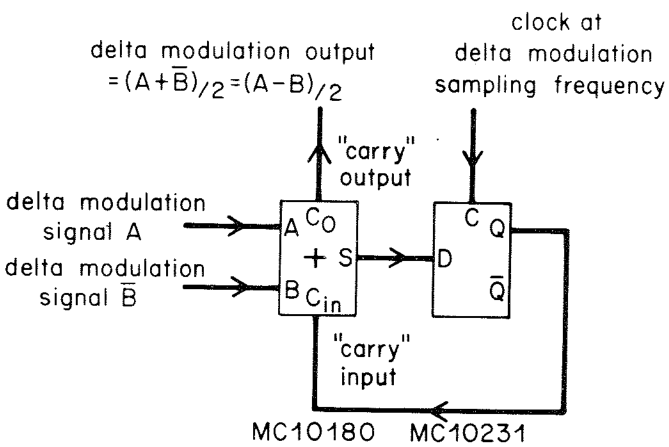
Fig. A1 - Circuit details of pseudo-random binary m-sequence generators



(a)



(b)



(c)

Fig. A2 - Circuit details of delta-modulation codec and subtractor
(*a*) Coder (*b*) Decoder
(*c*) Delta-modulation addition/subtractor

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